LISTING OF THE CLAIMS

1. (Previously Presented) A system in a network conferencing environment for delivering a plurality of video or audio signals, the system comprising:

a plurality of transmitters configured to transmit a set of data streams onto a network, wherein the set of data streams is generated from the plurality of video or audio signals and at least one of the transmitters includes a silence suppressor for removing silences or background from the data streams of the audio signals transmitted by the said at least one transmitter; and

at least one receiver for receiving the set of data streams from the network and recovering the data streams into audio or video signals, the receiver including a demultiplexer for dynamically selecting a subset of the set of data streams based on a source identifier and a payload type and two or more receiver payload handler modules and two or more corresponding decoder modules for handling and decoding two or more types of the data streams.

2. (Original) The system of claim 1 wherein one of the payload handler modules handles audio G.711 data and another handles audio G.723.1 data and one or more of the decoder modules decodes audio G.711 data and another decodes audio G.723.1 data.

4. (Previously Presented) The computer system of claim 2 wherein the demultiplexer is operatively coupled to the one or more decoders for routing data to one of the decoders based on the source identifier and the payload type.

5. (Original) The computer system of claim 1 further including an audio mixer operatively coupled to the two or more corresponding decoders.

6. (Original) The computer system of claim 1 further including a media rendering module operatively coupled to the one or more decoders.

7. (Original) The computer system of claim 1 wherein one or more of the payload handlers includes: means for reassembling or combining two or more data packets, means for reordering data packets, means for detecting and rejecting duplicate data packets, or means for computing and compensating delay jitter.

8. (Original) The computer system of claim 1 further including means for streaming data.

18. (Previously Presented) A method of conducting a network conference with two or more computer systems, the method comprising:

monitoring incoming audio or video data for each of a plurality of conference parties for active or inactive status;

monitoring incoming audio or video data for a new speaker;

replacing audio or video data having the inactive status with data for the new speaker;

receiving audio or video data from first and second computer systems;

determining the audio or video payload type for the audio or video data from the first computer system;

routing the audio or video data from the first computer system to a first decoder based on the determination of the audio or video payload type for the audio or video data and at least a first source identifier;

determining the audio or video payload type for the audio or video data from the second computer system; and

routing the audio or video data from the second computer system to a second decoder based on the determination of the audio or video payloadl type for the audio or video data and at least a second source identifier.

19. (Original) The method of claim 18 further comprising:

decoding the audio or video data from the first and second computer systems; and

rendering the audio or video data from the first and second computer systems.

20. (Original) The method of claim 19 wherein the audio or video data from the first or the second computer system is audio G.711 data, audio G.723.1 data, video H.261, or video H.263 data.

21. (Previously Presented) A network conferencing system comprising:

a real-time transport protocol (RTP) compliant demultiplexer that is adapted for:

receiving a plurality of RTP compliant data streams from a network;

dynamically selecting a portion of the RTP data streams;

routing one or more RTP data streams of the portion based on at least one payload type and at least one source identifier;

two or more receiver payload handler modules coupled to the demultiplexer for handling routed data streams;

two or more decoder modules coupled to the demultiplexer for decoding data; and

a rendering module coupled to the decoder for playing back one or more RTP data streams.

22. (Previously Presented) A machine readable medium comprising instructions for implementing the modules of claim 21.

23. (Original) A machine readable medium comprising instructions for implementing the method of claim 21.

24. (Previously Presented) A computerized conference system comprising:

receiving means for receiving, via a communications network, respective first and second sets of data of at least one payload type from respective first and second conference participants;

first and second decoder modules for respectively decoding the at least one payload type of data;

means for routing data received by receiving means to the first or the second decoder module based on the payload type and at least one source identifier;

means for determining whether one or more of the first and second sets of data is associated with an inactive conference participant; and

means, responsive to determination of the inactive conference participant, for substituting a third set of data from a third conference participant, for at least the one of the first and second sets of data associated with the inactive conference participant.

25. (Previously Presented) A method of operating a computerized conference system, comprising:

receiving, via a communications network, first and second audio data streams having at least two payload types from respective first and second conference participants;

decoding at least a portion of the first audio data stream in a first decoder for one of the at least two payload types of audio data the decoded portio of the first audio data stream determined by a first audio data stream source identifier and the payload type;

decoding at least a portion of the second audio data stream in a second decoder for a second of the at least two payload types of audio data the decoded portion of the second audio data stream determined by a second audio data stream source identifier and payload type;

determining whether one or more of the first and second audio data streams is associated with an inactive conference participant; and

substituting a third audio data stream for at least the one of the first and second audio data streams, the third audio data stream associated with the inactive conference participant.

26. (Previously Presented) A conference system for large numbers of participants, comprising:

means for receiving a plurality of audio data streams from a corresponding plurality of conference participants;

means for selecting a subset of the plurality of audio data streams, wherein the selected subset of audio data streams includes streams of different payload types;

decoder modules for decoding different payload types of audio data;

means for routing the selected subset of the plurality of audio data streams to the decoder modules based on the payload types of the streams and a plurality of source identifiers of the streams; and

means for rendering the selected subset of audio data streams.

28. (Original) The conference system of claim 26:

wherein the selected subset of audio data streams includes a first audio data stream and a second audio data stream; and

wherein the system further comprises:

means for determining whether one or more of the first and second audio data streams is associated with an inactive conference participant; and

means, responsive to determination of the inactive conference participant, for substituting a third audio data stream from a third conference participant, for at least the one of the first and second audio data streams associated with the inactive conference participant.

29. (Previously Presented) A conferencing method comprising:

receiving a plurality of audio data streams from a corresponding plurality of conference participants;

selecting a subset of the plurality of audio data streams, wherein the selected subset of audio data streams includes streams of different payload types;

routing the selected subset of the plurality of audio data streams to decoder modules based on their payload types and a plurality of source identifiers; and

rendering the selected subset of audio data streams.

31. (Original) The method of claim 29:

wherein the selected subset of audio data streams includes a first audio data stream and a second audio data stream; and

wherein the method further comprises:

determining whether one or more of the first and second audio data streams is associated with an inactive conference participant; and

substituting a third audio data stream from a third conference participant for at least the one of the first and second audio data streams associated with the inactive conference participant.

32. (Previously Presented) A conferencing method comprising:

receiving a plurality of data streams from a corresponding plurality of conference participants;

selecting a subset of the plurality of data streams, wherein the selected subset of audio data streams includes streams of different payload types;

routing the selected subset of the plurality of audio data streams to decoder modules based on their payload types and a plurality of source identifiers;

rendering the selected subset of data streams;

determining whether one or more of the first and second data streams is associated with an inactive conference participant; and

substituting a third data stream from a third conference participant, for at least the one of the first and second data streams determined to be associated with the inactive conference participant.

33. (Original) The method of claim 32, wherein the selected subset of data streams includes a first audio data stream formatted according to a first protocol and a second audio data stream formatted according to a second protocol.

34. (Original) The method of claim 32, wherein the selected subset of data streams includes a first video data stream formatted according to a first protocol and a second video data stream formatted according to a second protocol.

35. (Previously Presented) The system of claim 1, wherein the data streams in the selected subset are most recently activated data steams.

36. (Original) The system of claim 24, wherein the first and second sets of data are audio signal data.

37. (Previously Presented) The system of claim 1, wherein the source identifier is a synchronization source identifier (SSRC).

38. (Previously Presented) The network conferencing system of claim 21, wherein the source identifier is a synchronization source identifier (SSRC).

39. (Previously Presented) The computerized conference system of claim 24, wherein the source identifier is a synchronization source identifier (SSRC).

40. (Previously Presented) The method of claim 25 wherein the first audio data stream source identifier is a synchronization source identifier (SSRC); and the second audio data stream source identifier is a synchronization source identifier (SSRC).

41. (Previously Presented) The conference system of claim 26, wherein the plurality of source identifiers are synchronization source identifiers (SSRCs).

42. (Previously Presented) The method of claim 18, wherein the first source identifier and the second source identifier are synchronization source identifiers (SSRCs).